Version with Markings to Show Changes Made

8. (Amended) [The method of claim 1]

A method of reducing the amount of computations required to create a sound signal representing one or more sounds originating at a plurality of discrete positions in space, where the signal is to be perceived as simulating one or more sounds at one or more selected positions in space with respect to a listener, comprising the steps of:

(a) determining a spatial characteristic function for a position in space at which sound originating at a plurality of positions in space is to be received, wherein said characteristic function represents a head-related impulse response;

(b) applying said characteristic function as a filter to the signal representing sound to produce a filtered signal; and

(c) converting the filtered signal to a sound wave and producing the sound wave for a listener;

wherein the spatial characteristic function is determined for a selected number of N samples and a selected number of M eigen values and wherein the model filter function for an azimuth position θ and an elevation position φ of sound originating in a spherical coordinate system about the position of sound measurement as the origin has the form

$$y(n) = \sum_{m=1}^{M}\left[\sum_{k=1}^{K} w_m(\theta_k,\varphi_k)s_k(n)\right]q_m(n). \quad 9(c)$$

where s represents a sound source, K represents the number of independent sound sources, $w_m(\theta,\varphi)$ are the weighting factors, and $q_m(n)$ is a vector representing an orthonormal basis for a head-related impluse function.

16. (Twice Amended) An apparatus for providing sounds created by a plurality of sound sources to a listener which simulates the origin of each sound at a selected position in space with respect to the listener, comprising:

(a) an environment input for receiving information concerning a listening environment to be simulated and relative position of a listener;

(b) a calculator for receiving the information from said environment input, and calculating attenuation and time delays to simulate said environment and said listener position;

(c) a signal input for receiving a signal representing sound originating at a plurality of positions in space;

(d) a left channel and a right channel attached to said calculator and receiving said calculation of attenuation and time delay therefrom, and also attached to said signal input and receiving said sound signal from said signal input, each channel comprising:

(i) a source placement array for filtering said sound signal in accordance with a spatial characteristic function, wherein said spatial characteristic function is a head-related impulse response;

(ii) a[n] plurality of eigen filters attached to said source placement array and receiving the signal therefrom, wherein said eigen filters introduce time delays into said signal; and

(iii) a signal output for attaching a speaker to the apparatus, attached to said plurality of eigen filters for receiving and summing the signal therefrom.

Kindly add the claims as follows.

--21. (New) An apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space, each channel within said apparatus comprising:

at least one delayer for delaying a sound source signal;

at least one attenuator for attenuating a sound source signal;

a plurality of filters for filtering said attenuated sound signal;

a plurality of weighting elements to weight said filtered sound signals; and

a summer for summing said filtered sound signals;

wherein said plurality of filters remain constant, with at least one of said at least one delay element, said at least one attenuator, and said plurality of weighting elements adapted to change a perceptive position of said sound source signal to a listener.

22. (New) The plurality of sound sources according to claim 21, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

23. (New) A method for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space, each channel within said apparatus comprising:

delaying a sound source signal;

attenuating a sound source signal;

filtering said attenuated sound signal;

weighting said filtered sound signals; and

summing said filtered sound signals;

wherein said filtered attenuated sound signal remains constant, with at least one of said delayed sound source signal, said attenuated sound source signal, and said weighted filtered sound signals are adapted to change a perceptive position of said sound source signal to a listener.

24. (New) The plurality of sound sources according to claim 23, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

25. (New) An apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space, each channel within said apparatus comprising:

means for delaying a sound source signal;

means for attenuating a sound source signal;

means for filtering said attenuated sound signal;

means for weighting said filtered sound signals; and

means for summing said filtered sound signals;

wherein said means for filtering said attenuated sound signal remains constant, with at least one of said means for delaying said sound source signal, said means for attenuating said sound source signal, and said means for weighting said filtered sound signals are adaptived to change a perceptive position of said sound source signal to a listener.

26. (New) The plurality of sound sources according to claim 25, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

27. (New) An apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

a one source multiple reflection sound processor for processing a single sound source having multiple reflections;

a multiple source without reflections sound processor for processing multiple sound sources having no reflections; and

a multiple source multiple reflections sound processor for processing multiple sound sources having multiple reflections;

wherein said apparatus is adaptively able to switch on and off said one source multiple reflection sound processor, said multiple source without reflections sound processor and said multiple source multiple reflections sound processor.

28. (New) The apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space according to claim 27, further comprising:

a set of eigen filters that collectively represent the bases of at least one of head-related transfer functions and head-related impulse responses.

29. (New) A method for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

processing a single sound source having multiple reflections;

processing multiple sound sources having no reflections; and

processing multiple sound sources having multiple reflections;

wherein said processed single sound source having multiple reflections, said processed multiple sound sources without reflections and said multiple sound sources having multiple reflections are independently adaptively active.

30. (New) The method for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space according to claim 29, further comprising:

eigen filtering representing the bases of at least one of head-related transfer functions or head-related impulse responses.

31. (New) An apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

a single sound source having multiple reflections processing means for processing a single sound source having multiple reflections;

a multiple sound sources having no reflections processing means for processing multiple sound sources having no reflections; and

a multiple sound sources having multiple reflections processing means for processing multiple sound sources having multiple reflections;

wherein said means for processing a single sound source having multiple reflections, said means for processing multiple sound sources without reflections and said means for processing multiple sound sources having multiple reflections are independently adaptively activated.

32. (New) The apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space according to claim 31, further comprising:

eigen filtering representing the bases of at least one of head-related transfer functions or head-related impulse responses.

33. (New) An apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

a spatial feature extraction and regulation modeler for modeling a three dimensional sound;

wherein when a number of sound sources increases, a number of convolutions performed by said spatial feature extraction and regulation modeler is less than a multiple of a total number of sound sources.

34. (New) The plurality of sound signals according to claim 33 comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

35. (New) A method for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

modeling a three dimensional sound through a spatial feature extraction and regulation modeler; and

reducing a number of convolutions performed by said spatial feature extraction and regulation modeler by a value that is less than a multiple of a total number of sound sources when a number of sound sources increases.

36. (New) The plurality of sound signals according to claim 35, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

37. (New) An apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

modeling means for modeling a three dimensional sound through a spatial feature extraction and regulation modeler; and

reducing means for reducing a number of convolutions performed by said spatial feature extraction and regulation modeler by a value that is less than a multiple of a total number of sound sources when a number of sound sources increases.

38. (New) The plurality of sound signals according to claim 37, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

39. (New) An apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

a processor for processing said plurality of sound signals;

wherein an output of said processor is based on the functions:

$$\begin{aligned} y(n) &= s_1(n) * h(n,\theta_1,\varphi_1) + s_2(n) * h(n,\theta_2,\varphi_2) && (8a) \\ &= s_1(n) * \sum_{m=1}^{M} w_m(\theta_1,\varphi_1) q_m(n) + s_2(n) * \sum_{m=1}^{M} w_m(\theta_2,\varphi_2) q_m(n) && (8b) \\ &= \sum_{m=1}^{M} \left[w_m(\theta_1,\varphi_1) s_1(n) + w_m(\theta_2,\varphi_2) s_2(n) \right] * q_m(n) && (8c) \end{aligned}$$

where $h(n,\theta,\varphi)$ represents head-related impulse responses, $s_x(n)$ represents sound signals at different directions, $w_m(\theta_x,\varphi_x)$ represents weight functions, $q_m(n)$ represents eigen filters, and M represents the dimensions of a subspace.

40. (New) The plurality of sound signals according to claim 39, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

41. (New) The apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals according to claim 39, wherein:

said plurality of eigen filters is of a range from 3 to 16.

42. (New) A method for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

processing said plurality of sound signals;

outputting based on the functions:

$$\begin{aligned} y(n) &= s_1(n) * h(n,\theta_1,\varphi_1) + s_2(n) * h(n,\theta_2,\varphi_2) & (8a) \\ &= s_1(n) * \sum_{m=1}^{M} w_m(\theta_1,\varphi_1) q_m(n) + s_2(n) * \sum_{m=1}^{M} w_m(\theta_2,\varphi_2) q_m(n) & (8b) \\ &= \sum_{m=1}^{M} \left[w_m(\theta_1,\varphi_1) s_1(n) + w_m(\theta_2,\varphi_2) s_2(n) \right] * q_m(n) & (8c) \end{aligned}$$

where $h(n,\theta,\varphi)$ represents head-related impulse responses, $s_x(n)$ represents sound signals at different directions, $w_m(\theta_x,\varphi_x)$ represents weight functions, $q_m(n)$ represents eigen filters, and M represents the dimensions of a subspace.

43. (New) The plurality of sound signals according to claim 42, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

44. (New) The method for efficiently simultaneously processing a simulation of a plurality of sound signals, wherein:

said plurality of eigen filters is of a range from 3 to 16.

45. (New) An apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

processing means for processing said plurality of sound signals;

outputting means for outputting based on the functions:

$$\begin{aligned} y(n) &= s_1(n) * h(n,\theta_1,\varphi_1) + s_2(n) * h(n,\theta_2,\varphi_2) && (8a) \\ &= s_1(n) * \sum_{m=1}^{M} w_m(\theta_1,\varphi_1) q_m(n) + s_2(n) * \sum_{m=1}^{M} w_m(\theta_2,\varphi_2) q_m(n) && (8b) \\ &= \sum_{m=1}^{M} \left[w_m(\theta_1,\varphi_1) s_1(n) + w_m(\theta_2,\varphi_2) s_2(n) \right] * q_m(n) && (8c) \end{aligned}$$

where $h(n,\theta,\varphi)$ represents head-related impulse responses, $s_x(n)$ represents sound signals at different directions, $w_m(\theta_x,\varphi_x)$ represents weight functions, $q_m(n)$ represents eigen filters, and M represents the dimensions of a subspace.

46. (New) The plurality of sound signals according to claim 45, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

47. (New) The apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals according to claim 45, wherein:

said plurality of eigen filters is of a range from 3 to 16.

48. (New) An apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

a processor for processing said plurality of sound signals rendered from a one ear output signal which is the summation of each source convoluted with respective head-related impulse responses;

wherein an output of said processor is based on the functions:

$$\begin{aligned} y(n) &= s_1(n) * h(n,\theta_1,\varphi_1) + s_2(n) * h(n,\theta_2,\varphi_2) + \cdots + s_k(n) * h(n,\theta_k,\varphi_k) && (9a) \\ &= \sum_{k=1}^{K} s_k(n) * \sum_{m=1}^{M} w_m(\theta_k,\varphi_k) q_m(n) && (9b) \\ &= \sum_{m=1}^{M} \left[\sum_{k=1}^{K} w_m(\theta_k,\varphi_k) s_k(n) \right] * q_m(n). && (9c) \end{aligned}$$

where k represents independent sound sources at different spatial locations, $h(n,\theta,\varphi)$ represents head-related impulse responses, $s_x(n)$ represents sound signals at different directions, $w_m(\theta_x,\varphi_x)$ represents weight functions, $q_m(n)$ represents eigen filters, and M represents the dimensions of a subspace.

49. (New) The plurality of sound signals according to claim 48, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

50. (New) The apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals according to claim 48, wherein:

said plurality of eigen filters is of a range from 3 to 16.

51. (New) A method for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

processing said plurality of sound signals rendered from a one ear output signal which is the summation of each source convoluted with respective head-related impulse responses;

outputting based on the functions:

$$y(n) = s_1(n) * h(n,\theta_1,\varphi_1) + s_2(n) * h(n,\theta_2,\varphi_2) + \cdots + s_k(n) * h(n,\theta_k,\varphi_k) \quad (9a)$$

$$= \sum_{k=1}^{K} s_k(n) * \sum_{m=1}^{M} w_m(\theta_k,\varphi_k) q_m(n) \quad (9b)$$

$$= \sum_{m=1}^{M} \left[\sum_{k=1}^{K} w_m(\theta_k,\varphi_k) s_k(n) \right] * q_m(n). \quad (9c)$$

where k represents independent sound sources at different spatial locations, $h(n,\theta,\varphi)$ represents head-related impulse responses, $s_x(n)$ represents sound signals at different directions, $w_m(\theta_x,\varphi_x)$ represents weight functions, $q_m(n)$ represents eigen filters, and M represents the dimensions of a subspace.

52. (New) The apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals according to claim 51, wherein:

said plurality of eigen filters is of a range from 3 to 16.

53. (New) The plurality of sound signals according to claim 51, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

54. (New) An apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals in a three dimensional space comprising:

processing means for processing said plurality of sound signals rendered from a one ear output signal which is the summation of each source convoluted with respective head-related impulse responses;

outputting means for outputting based on the functions:

$$\begin{aligned} y(n) &= s_1(n) * h(n,\theta_1,\varphi_1) + s_2(n) * h(n,\theta_2,\varphi_2) + \cdots + s_k(n) * h(n,\theta_k,\varphi_k) && (9a) \\ &= \sum_{k=1}^{K} s_k(n) * \sum_{m=1}^{M} w_m(\theta_k,\varphi_k) q_m(n) && (9b) \\ &= \sum_{m=1}^{M} \left[\sum_{k=1}^{K} w_m(\theta_k,\varphi_k) s_k(n) \right] * q_m(n). && (9c) \end{aligned}$$

where k represents independent sound sources at different spatial locations, $h(n,\theta,\varphi)$ represents head-related impulse responses, $s_x(n)$ represents sound signals at different directions, $w_m(\theta_x,\varphi_x)$ represents weight functions, $q_m(n)$ represents eigen filters, and M represents the dimensions of a subspace.

55. (New) The apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals according to claim 54, wherein:

said plurality of eigen filters is of a range from 3 to 16.

56. (New) The plurality of sound signals according to claim 54, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

57. (New) An apparatus for efficiently processing a one source sound signal to simulate a three dimensional sound signal comprising:

a processor for processing said one source sound signal;

wherein an output of said processor is based on the functions:

$$\begin{aligned} y_L(n) &= s(n) * \sum_{m=1}^{M} w_m(\theta_L, \varphi_L) q_m(n), && (10) \\ &= \sum_{m=1}^{M} \left[w_m(\theta_L, \varphi_L) s(n) \right] * q_m(n), && (10a) \\ &= \sum_{m=1}^{M} \left[s(n) * q_m(n) \right] w_m(\theta_L, \varphi_L), && (10b) \end{aligned}$$

where, *s(n)* represents a sound signal, $w_m(\theta_x, \varphi_x)$ represents weight functions, $q_m(n)$ represents eigen filters, and M represents the dimensions of a subspace.

58. (New) The plurality of sound signals according to claim 57, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

59. (New) The apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals according to claim 58, wherein:

said plurality of eigen filters is of a range from 3 to 16.

60. (New) A method for efficiently processing a one source sound signal to simulate a three dimensional sound signal comprising:

processing said one source sound signal;

outputting based on the functions:

$$\begin{aligned} y_L(n) &= s(n) * \sum_{m=1}^{M} w_m(\theta_L, \varphi_L) q_m(n), && (10) \\ &= \sum_{m=1}^{M} \left[w_m(\theta_L, \varphi_L) s(n) \right] * q_m(n), && (10a) \\ &= \sum_{m=1}^{M} \left[s(n) * q_m(n) \right] w_m(\theta_L, \varphi_L), && (10b) \end{aligned}$$

where, $s(n)$ represents a sound signal, $w_m(\theta_x, \varphi_x)$ represents weight functions, $q_m(n)$ represents eigen filters, and M represents the dimensions of a subspace.

61. (New) The plurality of sound signals according to claim 57, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

62. (New) The method for efficiently simultaneously processing a simulation of a plurality of sound signals according to claim 60, wherein:

said plurality of eigen filters is of a range from 3 to 16.

63. (New) An apparatus for efficiently processing a one source sound signal to simulate a three dimensional sound signal comprising:

processing means for processing said one source sound signal;

outputting means for outputting based on the functions:

$$
\begin{aligned}
y_L(n) &= s(n) * \sum_{m=1}^{M} w_m(\theta_L, \varphi_L) q_m(n), && (10) \\
&= \sum_{m=1}^{M} \left[w_m(\theta_L, \varphi_L) s(n) \right] * q_m(n), && (10a) \\
&= \sum_{m=1}^{M} \left[s(n) * q_m(n) \right] w_m(\theta_L, \varphi_L), && (10b)
\end{aligned}
$$

where, $s(n)$ represents a sound signal, $w_m(\theta_x, \varphi_x)$ represents weight functions, $q_m(n)$ represents eigen filters, and M represents the dimensions of a subspace.

64. (New) The plurality of sound signals according to claim 63, comprising:

multiple reflections, multiple source without reflections and multiple source with multiple reflections.

65. (New) The apparatus for efficiently simultaneously processing a simulation of a plurality of sound signals according to claim 63, wherein:

said plurality of eigen filters is of a range from 3 to 16.

66. A method of producing a 3D sound with reduced computations with binaural or speaker presentations with which multiple independent sound sources and reflections of independent sound sources are summed together to create a synthesized 3D audio scene with improved speed and efficiency, comprising the following steps:

(a) determining a set of M eigen filters representing at least one of a measured head-related transfer functions (HRTFs) or head-related impulse responses HRIRs;

(b) determining a set of M spatial characteristics functions (SCFs) linearly combined with said eigen filters to reproduce at least one of said measured HRTFs, and said HRIRs;

(c) interpolating at least one of said HRTFs and said HRIRs in directions where a measurement was not made; and

(d) for each source, said multiple independent sound sources or reflections of independent sound sources, introducing at least one of a delay and a weight;

(e) three dimensionally positioning at least one of said multiple independent sound sources or reflections of independent sound sources by adapting said weight and said delay by a multiple, said multiple derived from sample values of spatial characteristic functions (SCFs) samples obtained through evaluating SCFs at an azimuth and an elevation intended for said source to be positioned. 2xM sub-signals are generated at the end of this step;

(f) for K independent and dependent sound sources, repeat steps (d) through (e) to generate 2xKxM sub-signals; and

(g) convolving said sub-signals with 2xM eigen filters to generate 2xM signals, said 2xM signals further regrouped into 2 binaural signals for a left ear and a right ear presentation.

67. (New) The method of producing a 3D sound according to claim 66, wherein:

said plurality of eigen filters is of a range from 3 to 16.

68. (New) An apparatus for producing a 3D sound with reduced computations with binaural or speaker presentations with which multiple independent sound sources and reflections of independent sound sources are summed together to create a synthesized 3D audio scene with improved speed and efficiency, comprising the following steps:

a first determiner determining a set of M eigen filters representing at least one of a measured head-related transfer functions (HRTFs) or head-related impulse responses HRIRs;

a second determiner determining a set of M spatial characteristics functions (SCFs) linearly combined with said eigen filters to reproduce at least one of said measured HRTFs, and said HRIRs;

an interpolator interpolating at least one of said HRTFs and said HRIRs in directions where a measurement was not made;

at least one of a delayer and a weighter for at least one of each source, said multiple independent sound sources and reflections of independent sound sources;

a positioner three dimensionally positioning at least one of said multiple independent sound sources or reflections of independent sound sources by adapting said weight and said delay by a multiple, said multiple derived from sample values of spatial characteristic functions (SCFs) samples obtained through evaluating SCFs at an azimuth and an elevation intended for said source to be positioned. 2xM sub-signals are generated at the end of this step;

a repeater repeated utilization for K independent and dependent sound sources said delayer and weighter and said positioner to generate 2xKxM sub-signals; and

a convolver convolving said sub-signals with 2xM eigen filters to generate 2xM signals, said 2xM signals further regrouped into 2 binaural signals for a left ear and a right ear presentation.

69. (New) The apparatus for producing a 3D sound according to claim 68, wherein:

said plurality of eigen filters is of a range from 3 to 16.

70. (New) An apparatus for producing a 3D sound with reduced computations with binaural or speaker presentations with which multiple independent sound sources and reflections of independent sound sources are summed together to create a synthesized 3D audio scene with improved speed and efficiency, comprising the following steps:

a first determiner means for determining a set of M eigen filters representing at least one of a measured head-related transfer functions (HRTFs) or head-related impulse responses HRIRs;

a second determiner means for determining a set of M spatial characteristics functions (SCFs) linearly combined with said eigen filters to reproduce at least one of said measured HRTFs, and said HRIRs;

an interpolator means for interpolating at least one of said HRTFs and said HRIRs in directions where a measurement was not made;

at least one of a delayer means and a weighter means for at least one of each source, said multiple independent sound sources and reflections of independent sound sources;

a positioner means for three dimensionally positioning at least one of said multiple independent sound sources or reflections of independent sound sources by adapting said weight and said delay by a multiple, said multiple derived from sample values of spatial characteristic functions (SCFs) samples obtained through evaluating SCFs at an azimuth and an elevation intended for said source to be positioned. 2xM sub-signals are generated at the end of this step;

a repeater means for repeated utilization for K independent and dependent sound sources said delayer means and weighter means and said positioner means to generate 2xKxM sub-signals; and

a convolver means for convolving said sub-signals with 2xM eigen filters to generate 2xM signals, said 2xM signals further regrouped into 2 binaural signals for a left ear and a right ear presentation.

71. (New) The apparatus for producing a 3D sound according to claim 70, wherein:

said plurality of eigen filters is of a range from 3 to 16.

72. (New) An apparatus producing a three dimensional sound comprising:

a set of M eigen filters derived from measuring at least one of a head related transfer functions (HRTFs) or head related impulse response (HRIRs) with independence of location of a source;

a set of M delays, multipliers, and spatial characteristic function (SCF) weights;

a set of M combiners;

a summer and regrouper, one said summer and said regrouper for each sound of a sound destination.

73. (New) The apparatus for producing a 3D sound according to claim 72, wherein:

said plurality of eigen filters is of a range from 3 to 16.

74. (New) A method for producing a three dimensional sound comprising:

deriving a set of M eigen filters from measuring at least one of a head related transfer functions (HRTFs) or head related impulse response (HRIRs) with independence of location of a source;

setting M delays, multipliers, and spatial characteristic function (SCF) weights;

combining a set of M sounds;

summing and regrouping, summing and regrouping for each sound of a sound destination.

75. (New) The method of producing a 3D sound according to claim 74, wherein:

said plurality of eigen filters is of a range from 3 to 16.

76. (New) A method for producing a three dimensional sound comprising:

deriving means for deriving a set of M eigen filters from measuring at least one of a head related transfer functions (HRTFs) or head related impulse response (HRIRs) with independence of location of a source;

setting means for setting M delays, multipliers, and spatial characteristic function (SCF) weights;

combining means for combining a set of M sounds;

summing means and regrouping means for each sound of a sound destination.

77. (New) The apparatus for producing a 3D sound according to claim 68, wherein:

said plurality of eigen filters is of a range from 3 to 16.--

REMARKS

Claims 8 and 16 are amended herein. Claims 21-77 are added herein. Claims 1-77 now remain pending in the application.

The Applicants respectfully request the Examiner to reconsider his earlier rejections in light of the following remarks. No new issues are raised nor is further search required as a result of the changes made herein. Entry of the Amendment is respectfully requested.

Allowable Claim

The Applicant thanks the Examiner for the indication that claim 8 recites allowable subject matter. Claim 8 is amended herein to be in independent form. Claim 8 is now in condition for allowance.

Claims 1-7, 9-16 and 18 over Chen and Nagamitsu

In the Office Action, claims 1-7 were rejected under 35 U.S.C. §103(a) as allegedly being obvious over Chen et al. U.S. Patent No. 5,500,900 ("Chen") in view of Nagamitsu et al. U.S. Patent No. 5,467,401 ("Nagamitsu"), with claims 9-16 and 18 rejected as allegedly obvious over Chen. The Applicant respectfully traverses the rejection.

Claims 1-7 recite, *inter alia,* applying a spatial characteristic function that represents a head-related impulse response as a filter to a signal representing sound to produce a filtered signal. Claims 9-15 recite, *inter alia*, a filter comprising a linear function including a spatial component which comprises a head-related impulse response. Claims 16 and 18 recite, *inter alia*, a source placement array for filtering a sound signal in accordance with a spatial characteristic function linearly combined with an eigen filter to form a head-related impulse response.

Chen teaches a free-field-to-eardrum transfer function (FETF, an previous name for an HRTF) developed by comparing auditory data for points in three-dimensional space for a model ear and auditory data collected for the same listening location with a microphone (Abstract). Each FETF is represented as a weighted sum of frequency-dependent functions obtained from an expansion of a

measured FEFT covariance matrix (Chen, Abstract). Spatial transformation characteristic functions (STCF) are applied to transform the weighted frequency-dependent factors to functions of spatial variables for azimuth and elevation (Chen, Abstract). A generalized spline model is fit to each STCF to filter out noise and permit interpolation of the STCF between measured points (Chen, Abstract). A spline model used to generate the STCFs, smooths measurement noise and enables interpolation of the STCFs between measurement directions (Chen, col. 5, lines 18-20). A regularizing parameter within the spline model controls a trade-off between smoothness of a solution and its fidelity to the data (Chen, col. 5, lines 29-31).

The Office Action relies on Nagamitsu to allegedly make up for the deficiencies in Chen to arrive at the claimed invention. The Applicant respectfully disagrees.

Nagamitsu appears to teach a sound environment simulator including a sound field analyzing unit, a sound field reproducing unit, and an output unit (Abstract). The sound environment analyzing unit divides the solid surfaces of a space to be analyzed into a set of sections to compute a volume with a sound absorption coefficient of walls and form factors (Nagamitsu, Abstract). Time series data relates to the arrival volume of sounds emanated from a certain sound source to a sound receiving point (Nagamitsu, Abstract). An impulse response computing unit in the sound field reproducing unit transduces the time series data into an impulse response (Nagamitsu, Abstract). A sound receiving point receives two types of sounds, a direct sound from the sound source and a reflected sound (Nagamitsu, col. 6, lines 39-42).

Chen teaches use of a head related transfer function within the background of the invention. A <u>head related transfer function</u> is <u>**NOT**</u> a <u>head-related impulse response</u>. Chen fails to teach processing multiple sound sources efficiently to generate three dimensional sounds with multiple sources based on a <u>head-related impulse response</u>, much less <u>filtering</u> based on a <u>head-related impulse response</u>, as claimed by claims 1-7, 9-16 and 18.

Nagamitsu fails to teach a spatial function that represents a <u>head-related impulse response</u>.

Neither Chen nor Nagamitsu, either alone or in combination, disclose, teach or suggest filtering based on a head-related impulse response, as claimed by claims 1-7, 9-16 and 18.

Accordingly, for at least all the above reasons, claims 1-7, 9-16 and 18 are patentable over the prior art of record. It is therefore respectfully requested that the rejection be withdrawn.

Claim 20 over Chen in view of Sekine

In the Office Action, claim 20 was rejected under 35 U.S.C. §103(a) as allegedly being obvious over Chen in view of Sekine et al. U.S. Patent No. 5,822,438 ("Sekine"). The Applicant respectfully traverses the rejection.

Claim 20 is dependent on claim 16, and is allowable for at least the same reasons as claim 16.

Claim 20 recites, *inter alia*, a source placement array for filtering a sound signal in accordance with a spatial characteristic function, wherein the spatial characteristic function is a head-related impulse response.

As discussed above, Chen fails to teach a head-related impulse response, much less filtering based on a head-related impulse response, as claimed by claim 20.

The Office Action relies on Sekine to allegedly make up for the deficiencies in Chen to arrive at the claimed invention. The Applicant respectfully disagrees.

Sekine appears to teach an electronic musical instrument having a signal mixing portion and a virtual-speaker position control portion (Sekine, Abstract). The virtual-speaker position control portion applies different delays times to each of a plural mixed signals to output delayed signals to left and right speakers (Sekine, Abstract). A cross talk canceler is used to cancel cross-talk sounds which emerge when a person hears sounds with both ears (Sekine, col. 5, lines 21-23).

Sekine fails to teach a spatial function that represents a head-related impulse response.

Neither Chen nor Sekine, either alone or in combination, disclose, teach or suggest filtering based on a head-related impulse response, as claimed by claim 20.

Accordingly, for at least all the above reasons, claim 20 is patentable over the prior art of record. It is therefore respectfully requested that the rejection be withdrawn.

Conclusion

All objections and rejections having been addressed, it is respectfully submitted that the subject application is in condition for allowance and a Notice to that effect is earnestly solicited.

Respectfully submitted,

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